important backbones of data communication and transmission. Email, file access and sharing, and services access and sharing are but a few of the many data communication service[s] and applications provided by such networks. Recently, next generation data communication applications such as Voice over IP (VoIP) and real-time interactive multi-media have also begun to emerge.

On page 3, please replace the paragraph beginning on line 6 with the following paragraph:

Packets lost during transmission also adversely impact the quality of VoIP communications. It has been demonstrated that speech becomes unintelligible if voice packets comprising more than about 60ms of digitized speech data are lost. Packets can be lost in transmission for three reasons: (1) if the electrical signal suffers from an electromagnetic disturbance, thus causing an error in one or more bits or (2) if the queue it is waiting in for transmission at some intermediary router along the path overflows due to congestion, thus causing packet dropping or (3) because of some configuration errors that cause packet transmission collisions (i.e., two or more electrical transmission signals overlapping and jamming each other). Because VoIP is a real-time interactive data communications application the current Internet protocols that provide for retransmission are of little help in this instance since late packets may become outdated, i.e., useless, when the packets finally arrive.

On page 3, please replace the paragraph beginning on line 19 with the following paragraph:

Packet jitter also substantially affects the quality of VoIP communications. In VoIP, packet jitter may result in the inability to reassemble all packets within time limits necessary to meet minimum acceptable latency requirements. As a consequence, sound quality can suffer due to the absence of some packets in the reassembly process, i.e., loss of some voice data or excessive packet delay. It has been determined that to achieve acceptable voice quality, voice packet inter-arrival times (i.e., jitter) generally must be limited to about 50-75msec. Within this range, data buffering can be used to smooth out jitter problems without substantially affecting the overall quality of the voice communications.

On page 4, please replace the paragraph beginning on line 3 with the following paragraph:

Additionally, the current Internet addressing and routing protocols and approaches for fixed node data networks are incapable of supporting the dynamically changing addressing and routing situations that arise in recently proposed wireless, mobile-access digital data networks. The International Telecommunication Union (ITU) of the Internet Society, the recognized authority for worldwide data network standards, has recently published its International Mobile Communications-2000 (IMT-2000) standards. These standards propose so-called third generation (3G) and beyond (i.e., 3.5G, 4G, etc.) data networks that include extensive mobile Internet access by wireless, mobile node devices including cellular phones, personal digital assistants (PDA's), handheld computers, and the like. (See http://www.itu.int).

On page 5, please replace the paragraph beginning on line 9 with the following paragraph:

With circuit-switched core networks in current wireless communication systems, measurements of layer 2 QoS parameters over the wireless link only is sufficient to decide <u>to</u> which access point to handoff the wireless communication device. This is because the circuit-switched core network is robust and well provisioned to provide reliable and stable service. In that case, the wireless link is the only bottleneck in the end-to-end path and it is appropriate to base the handoff trigger on QoS measurements on that portion of the communication path.

On page 15, please replace the paragraph beginning on line 13 with the following paragraph:

In the preferred embodiment of the present invention, the weights [$\alpha_{,j}$] $\underline{\alpha_{i,j}}$ depend on the real-time application under consideration, for example, in voice delay is more important than bandwidth, etc. These weights will give a measure of how much each parameter is more important than the rest. We have: $0 \leq \alpha_i \leq 1$. And for voice we have: $0 \leq QoS\_Quantifier_i \leq 4$.